Table 4-2. File names for voice announcements (Page 2 of 2)

Call state	Announcement purpose	Announcement filename
Gateway not available	Used when a destination gateway is not able to accept the call at the present time. This is currently used for: • A gateway is not registered with the gatekeeper. • A gateway can not handle current call load/rate.	h323ngw.au
Call failure	Default call failure prompt.	h323f.au
Call drop request (gatekeeper initiated)	The gatekeeper sent a request to drop an active call (DRQ) to the gateway. This is used in conjunction with debt-account billing systems.	h323drq.au

Enabling voice announcements

The TAOS unit supports play out of voice announcements for both IPDC and H.323 VoIP calls. Announcements are enabled by

- Specifying announcement files in STN messages from the SS7 signaling gateway for IPDC
- Enabling the use of voice announcements from within the `voip { 0 0 }` profile for H.323 call operations

Note By default, both IPDC and H.323 VoIP voice announcement file names are the same. The IPDC STN messages from the SS7 signaling gateway must call voice announcements using file names stored on the TAOS unit .

Enabling voice announcements for IPDC calls

For IPDC VoIP, voice announcements are requested by the SS7 signaling gateway. A request to play an announcement is passed to the TAOS unit, in IPDC format. From the IPDC specification (*Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15)*, announcements are specified as Send Tones or Announcement (STN) messages with the following tags:

- Tone Type (0x49) (now supports a value of "0x03 - Voice Announcement")
- Announcement Treatment (0x86) (number of times to play the announcement)
- Tone String (0x33) (the file name or identifier of the announcement to be played)

Call-progress tone parameters set in the `voip` profiles have no affect on IPDC announcement playbacks.

Note This feature requires obtaining a pre-paid billing application.

STN Message

Voice announcement playlists allow a list of announcement files to be signaled to the MultiVoice Gateway in the STN message. The following describes how tags in the STN message support voice announcements.

Tag 0x33 (Tone String)

This tag accepts the name of an announcement file, or a comma-separated list of announcement files (for example, h323dns.au, 1.au, 2.au). Intervening blanks are optional.

In addition, a playlist format is supported. This format is:

```
(c,d,(filename,c,d)...(filename,c,d))
```

where:

```
c = playCount (default = 1)
d = delay (default = 0) in milliseconds.
```

An example is:

```
(1,0(file1.au,1,5)(file2.au,2,5)(file3.au,1,5))
```

This format is useful if you want to specify non-default playcount and delay values for individual files in the playlist, since there is no way to signal this through IPDC.

Tag 0x86 (Announcement Treatment)

If a list of announcement files has been specified, the value of this tag is applied to the entire list in sequence. For example, if the value is 2, the entire list is played twice.

If a playlist format is used, this value is ignored.

An example of where Tag 0x86 is equivalent to 2 is:

```
Tag 0x33 = h323f.au,h323dns.au
Tag 0x86 = 2
```

is equivalent to:

```
Tag 0x33 = (2,0,(h323f.au,1,0),(h323dns.au,1,0))
```

Break-in voice announcements in IPDC

You can configure a voice announcement to be played while a packet call is in progress. While a break-in voice announcement is playing, the Real-time Transport Protocol (RTP) flow to the called party is suspended, the calling party hears the voice announcement, and the called party hears silence.

Typically, a break-in announcement is played to the calling party, however it could be played to the called party or both parties.

Note Break-in voice announcements are supported for packet calls, but not for time-division multiplexing (TDM) calls.

STN/ATN message call flow

Break-in voice announcements utilize existing STN/ASTN messaging as follows, fully supporting the cancel operation. Refer to *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15* for a description of messages (for example, RTP, STN, etc.).

```
                 TAOS Unit                TAOS unit
                 (incoming)   Softswitch  (outgoing)
                 ----------   ----------  ----------
                      |            |           |
                      |    RTP     |           |
                      |<-------------------------->|  Voice exchange in
                      |            |           |   progress
                      |    STN     |           |
                      |<-----------|           |   Break-in request
                      |    ASTN    |           |
                      |----------->|           |   Break-in started
break-in              |            |           |
announcement          |            |           |
is heard by           |            |           |
calling party         |            |           |
          .                 .            .
          .                 .            .
          .                 .            .
                      |    ASTN    |           |
                      |----------->|           |  Break-in completed
                      |    RTP     |           |
                      |<-------------------------->|  Voice exchange
                      |            |           |  resumes
                      |            |           |
```

Break-in announcement call flow — call release

If a break-in announcement is playing and an RCR is received, the call is released and an ACR is sent containing the RTP statistics for the packet call:

```
                 TAOS Unit                TAOS Unit
                 (incoming)   Softswitch  (outgoing)
                 ----------   ----------  ----------
                      |            |           |
                      |    RTP     |           |
                      |<-------------------------->|Voice exchange in
                        progress
                      |    STN     |           |
                      |<-----------|           | Break-in request
                      |    ASTN    |           |
                      |----------->|           | Break-in started
     break-in         |            |           |
     announcement     |            |           |
     is heard by      |            |           |
     calling party    |            |           |
              .            .            .
              .            .            .
```


S 1502

```
|     RCR      |                 |
|<-------------|                 |Release packet call
|     ACR      |                 |
|------------->|                 |Packet call released
```

Enabling voice announcements for H.323 calls

Three parameters located in the voip { 0 0 } profile are used to enable MultiVoice voice announcements for H.323 call processing:

Parameter	Specifies
h323-voice-ann-enabled	Enable voice announcement play out. Setting the parameter to yes enables voice announcement play out; No disables this feature. The default value for this parameter is no. Changes to this parameter are effective with the next VoIP call. Even when voice announcements are enabled, users still hear call-progress tones under the following conditions: • A traditional call progress tone is available (ringback, busy, etc.). • The gateway cannot play any more simultaneous announcements.
voice-ann-dir	The directory location where the voice announcement files are stored on the TAOS unit. This value defaults to the /current directory on pc-flash card 1. Changes to this parameter are effective with the next VoIP call. Enter the file path to the directory location containing the H.323 VoIP voice announcement files. This may be a string of 40 characters or less beginning with "/". This parameter defaults to N/A when h323-voice-ann-enabled = no. Changes to this parameter are effective with the next VoIP call.
voice-ann-enc	Either the G.711 μ-law or G.729 encoding of voice announcements that are played out by a MultiVoice Gateway. Voice announcements are used for reporting call progress to callers.

These parameters have no effect on SS7-IPDC voice announcement play out.

To enable voice announcements for H.323 call operations on a TAOS unit:

1 Open the voip { 0 0 } profile.

```
admin> read voip { 0 0 }
VOIP/{0 0} read
```

2 Turn on voice announcement play out by setting the h323-voice-ann-enable parameter to yes. For example:

```
admin> list h323-voice-ann-enabled
[in VOIP/{ 0 0 }:h323-voice-ann-enable]
h323-voice-ann-enable = no

admin> set h323-voice-ann-enabled = yes
```

3 Write your changes to the voip { 0 0 } profile.

```
admin> write voip { 0 0 }
VOIP/{0 0} written
```

Enabling G.711 μ-Law or G.729 encoding

Either G.729 or G.711 encoded speech for voice announcement play out can be used on MultiVoice Gateways. You configure the choice through the TAOS administration interface.Audio encoded in G.729 format is eight times smaller than the audio encoded in G.711, which allows customers to store and play a larger number of voice announcements.

Note While, G.711 encoded voice announcements files can be created with standard off-the-shelf software, a special tool is need to create G.729 encoded voice announcements that the MultiVoice Gateway will recognize and be able to play. This tool is available free to customers from the Lucent Technologies FTP download site. Customers requiring this tool should contact their account representative for details.

You configure the voice-ann-enc parameter to the voip profile on the MultiVoice Gateway as illustrated in the following example:

```
[in VOIP/{ 0 0 }]
voip-index* = { 0 0 }
gatekeeper-ip = 135.92.52.138
gk-mlg-control = no
......
voice-ann-enc = g711-ulaw
......
```

The voice-ann-enc parameter specifies either the G.711 μ–law or G.729 encoding of voice announcements that are played out by a MultiVoice Gateway. Voice announcements report call progress to callers. Changes to the voice-ann-enc parameter are effective with the next VoIP call.

The following example illustrates how to configure a MultiVoice Gateway to use G.729 encoding for voice announcement play out.

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set voice-ann-enc=g729

admin> write
```

The voice-ann-enc parameter has the following dependencies:

- The MultiVoice Gateway must be configured for using voice announcements to report call progress.
- Before a MultiVoice Gateway is configured to use G.729 voice-announcement encoding (voice-ann-enc=g729), voice announcement files must be converted to the G.729 compatible format. Lucent Technologies offers a tool, at no charge to MultiVoice Customers, that creates G.729 encoded voice announcement files.

- The MultiVoice Gateway must be configured to use G.729 voice-announcement encoding (`voice-ann-enc=g729`) when the Lucent Technologies prepaid billing message set is used for reporting call progress and for billing announcements.

Creating voice announcements

To create or edit a voice announcement, proceed as follows:

1. **Using your favorite midi or audio editor on your PC, open the file containing the voice announcement.**
2. **Create or edit the contents of the message. Then save your work.**
3. **Change the recorded file to an `.au` formatted file.**
4. **Move the file to a TFTP server download directory (for example, `C:\Users\default`)**
5. **From the TAOS unit, enter the CLI Load command using the load type `file`, as in the following example:**

   `load file network xxx.xxx.xxx.xxx filename.au`

   where `xxx.xxx.xxx.xxx` is the IP address of the TFTP server. By default, this command writes to the destination directory /current on flashcard 1.

Once the file is loaded on the flash card, the message is available for playback.

To make use of voice announcements, the flash card must use the FAT 12 format. Use the CLI Format command to apply the FAT format to the flash card.

The following dependencies apply when planning and creating voice announcements:

- Voice announcement files must always be in *.au* format. There are easily accessible tools available to convert from the standard Windows *.wav* format to an *.au* format file, including
  - CoolEdit (`http://www.syntrillium.com/cooledit/index.html`)
  - GoldWave32 (`http://www.goldwave.com`)
- The number of announcements that can be stored on a TAOS unit is limited only by the available space on the flash card, which has a capacity of 32Mb for the APX or 8Mb for the MAX TNT. However, having many different announcements might impact performance and sound quality during playback because the cache size is limited.

Formatting flash cards

Before using a PCMCIA card for loading voice announcement files, you must format it using a FAT format. The Format command creates this DOS-compatible flat file system by default on PCMCIA flash cards.

After inserting a flash card into slot 1 in the Shelf-controller, enter the Format command, as in the following example:
admin> **format flash-card-1**

or
admin> **format 1**

Before loading any tar images or message files, use the fsck command to verify the format, as in the following example:

```
admin> fsck
Volume Stats:
        Block Size: 512 (typical: 512)
    Blocks Per Cluster: 3 (typical: 1, may be powers of 2 up to 16)
    Reserved Blocks: 1 (typical: 1, but may be 0 - hundreds)
    Number of FATs: 2 (must be 2)
    Number of Root Directory Entries: 96 (typically between 32 and 224)
    Total Blocks: 11264
    Media Descriptor: f0 (ignored)
Volume Info calculated from values above:
    Blocks Per Fat: 11
    Fat Start Block: 1
    Root Dir Start Block: 23
    Data Start Block: 29
    Number of Root Dir Blocks: 6
    Number of Clusters: 3745
    FAT Type: Fat12
Cluster Usage
    Usable Clusters: 3743
    Free Clusters: 1828
    Clusters lost during interrupted writes: 0
    Other reserved clusters: 1909
admin>
```

For new format cards, fsck prints a summary of the file structure on the card (parenthetical text is not part of the output of this command).

Creating the voice announcements directory

You can create up to four directories on the external flash memory card for customized voice announcements. These voice announcements can be used to report call progress or for playback on command from MVAM, Softswitch, or a third-party billing application.

After creating any directory on a flash card and moving voice announcement files into it, specify the location by entering a pathname in the voice-ann-dir setting. For example, the following commands create a directory named messages and a subdirectory named announce on the flash card in slot 1:

```
admin> mkdir 1/messages
admin> mkdir 1/messages/announce
```

The following command loads a voice-announcement file named busy.au from a TFTP server at 10.10.10.10 to the /current directory on flash card 1 (flash card 1 is the default):

```
admin> load file network 10.10.10.10 busy.au
```

The following command moves the busy.au file to the new subdirectory on flash card 1:

```
admin> mv 1/current/busy.au 1/messages/announce/busy.au
```

The following commands inform the MultiVoice subsystem of the location of the voice announcement files:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set voice-ann-dir = /messages/announce
```


S 1506

```
admin> write
VOIP/{ 0 0 } written
```

You can specify a pathname up to 40 characters long. When the system receives a request to play an announcement, it looks in the specified directory on the flash card in slot 1. If the card is not present or the voice announcement file is not found, the system looks for the specified directory on flash card 2.

Displaying voice announcement files

The voice announcement files are displayed by name with all the other code images and tarballs stored on the flash card, when you enter the dircode command. For example:

```
admin> dircode
Flash card code directory:
Card 1, format FTL/FAT, capacity 8MB
/current:
    shelf-controller1231877Tue Oct 27 17:17:22 1998 7.1.0
    8t1-card209191Tue Oct 27 17:17:42 1998 7.1.0
    4ether-card180385Tue Oct 27 17:17:56 1998 7.1.0
    hdlc-card588610Tue Oct 27 17:18:38 1998 7.1.0
    48modem-56k-card724319Tue Oct 27 17:19:58 1998 7.1.0
    hold.au72723Thu Jan 21 19:54:12 1999
admin>
```

The dircode command output includes the type and capacity of each card inserted. For an old-format card, the output is unchanged from previous versions. For FAT-format cards, the output is slightly different, since information about each image is stored differently.

Voice announcement log messages

Several log messages are generated to alert the user of invalid voice announcement files. These include the following:

Table 4-3. Error/log messages (Page 1 of 2)

Message	Cause
LOG error, Shelf 1, Controller, Time: 19:06:23-- Announcement File /current/test.au not found	Appears when the TAOS unit cannot find the requested announcement.
LOG error, Shelf 1, Controller, Time: 20:32:07-- Invalid voice announcement format for drip.au	Appears when the voice announcement file does use a valid *.au* format file

Table 4-3. Error/log messages (Page 2 of 2)

Message	Cause
LOG error, Shelf 1, Controller, Time: 15:00:11-- Au File vopc4.a not supported, format 27, samplingRate 44100, channelCount 2	Appears when the voice announcement file uses a valid .au format file but does not meet the attributes for sampling rate, data format, channel count, string length, or file size.
LOG error, Shelf 1, Controller, Time: 15:00:11-- Announcement file too large. Max size = 102400	Appears when the voice announcement audio size is larger than 200K.

WARNING messages

Table 4-4. Warning messages

Message	Cause
LOG error, Shelf 1, Controller, Time: 15:00:20-- Voice Announcement info string in header is too big.	Appears when the information string of the .au file header is larger than can be handled. This is not harmful. Announcement playback still succeeds.

MultiVoice Real-time Fax

5

Real-time fax processing

Real-time fax calls begin when a VoIP call is placed from an originating fax machine to the answering machine. If the TAOS unit is configured to perform out-of-band dual tone multi-frequency (DTMF) signaling, a DSP automatically enables inband DTMF signaling at the start of the fax call. When the destination fax machine picks up the call and sends an answer tone, known as a CED tone, the destination Gateway detects this tone and initiates a switchover to real-time fax on both itself and the Gateway at the other end of the call. When the switchover is complete, the fax transmission proceeds normally.

The switch over from voice session to fax can occur at any point in a call. For example, the call can start as a normal VoIP call where two participants converse for any length of time before initiating a fax. Once the switch to fax is made, however, there is no switching back; i.e. once in fax mode, the fax is sent and then the call will be terminated.

You must create the appropriate coverage areas on the MultiVoice Access Manager to ensure that fax calls are routed between Gateways that are fax capable. For details, see the *MultiVoice Access Manager User's Guide.*

Note For MultiVoice to complete a fax call, both the Gateways involved in the call must have real-time fax enabled.

H.323 Annex D T.38 fax support

Support for real-time facsimile (FAX) interoperability with other vendors H.323 gateways is made possible through the implementation of the H.323 Annex D standard.

This implementation, which also requires the use of recommendation T.38, allows FAX interoperability with gateways of other vendors that have implemented this standard. MultiVoice-to-MultiVoice communication for fax uses a prestandard version of T.38, which includes improves feature performance.

MultiVoice gateways automatically detect other gateways of a call that are not MultiVoice gateways. Subsequently, if fax tones are detected, this system uses the H.323 Annex D standard for fax. MultiVoice gateways identify themselves in the `vendorIdentifier` fields of H.225 call signaling messages.

This feature implements section D.5 entitled "Replacing an existing audio stream with a T.38 fax stream" of the H.323 Annex D standard, version 4 dated November, 2000. The implementation of this feature also uses nonStandardParameter fields to indicate T.38 support in H.245 messages requiring an indication for T.38.

Real-time fax configuration

MultiVoice real-time fax is an implementation of the ITU-T T.38 standard for fax transmission across IP networks, using the VoIP framework for call establishment, fax initiation, and detection of an incoming fax call.

Note Real-time fax communications require guaranteed quality of service between the two fax-capable gateways. The packet loss on the network must be less than 1%.

To use MultiVoice real-time fax:

- The network must use TAOS units.
- The TAOS units must be licensed to provide support for the real-time fax feature.

Configure the TAOS unit to process real time fax calls by enabling fax mode through the `voip {0 0}` profile. By default, all calls begin as voice calls. Upon detection of a fax tone, the call type switches to fax mode.

Base profiles

For MultiVoice real-time fax to be available, the TAOS unit must be licensed to support both real-time fax and VoIP call processing. The base profile must include the following parameters:

Table 5-1. Base profile parameters

Parameter	Description
`voip-enabled`	Enables/disables VoIP call processing. When this parameter is set to yes, the TAOS unit has been hashed to process VoIP calls.
`rtfax-enabled`	Enables/disables real-time fax processing on a TAOS unit for H.323 call operations. When this parameter is set to yes, the TAOS unit has been licensed to process faxes for H.323 calls.

Rt-fax-options sub-profile

Following are the `rt-fax-options` sub-profile parameters (shown with default values) for enabling and improving the performance of real-time fax processing. Changes to these parameters take effect with the next VoIP call.

```
[in VOIP/{ 0 0 }:rt-fax-options]
rt-fax-enable = no
```


S 1510

```
ecm-enable = yes
low-latency-mode = yes
command-spoof = yes
local-retransmit-lsf = yes
packet-redundancy = 0
fixed_packets = no
max-data-rate = 14400
```

Parameter	**Setting**
rt-fax-enable	Enable/disable real-time fax call processing. When the parameter is set to no (the default), fax tones are passed as if they were normal voice samples, and the other parameters in the sub-profile are not applicable. When the parameter value is set to yes, the TAOS unit switches over from voice session to fax upon detection of a CED tone or V.21 HDLC flag.
ecm-enable	Enable/disable error correction mode (ECM) for real-time fax calls. When the parameter is set to yes (the default), fax frames can be retransmitted in the event that a frame is not received correctly. ECM frames are relayed end to end between terminals. Setting the parameter to no disables ECM, so fax frames containing errors are not corrected.
low-latency-mode	Enable/disable low latency mode for real-time fax operations over networks with low packet loss and low latency characteristics. Low latency mode allows operation on networks with less than 2.5 seconds or less of aggregate latency between pages. When the parameter is set to no, a minimum of 10 seconds delay is added to processing fax calls to allow interpretation of T.30 frames and implement spoofing.
command-spoof	Enable/disable spoofing of certain fax commands. Command spoofing is a method of improving performance and reducing fax errors on low latency networks.
local-retransmit-lsf	Enable/disable local retransmission of a low speed fax frame if no response is detected from the destination fax. This is designed to reduce fax transmission errors on low packet loss networks.
packet-redundancy	Improves reliability of MultiVoice real-time fax transmissions. Packet redundancy is recommended when transmissions are sent over unmanaged networks (such as the public Internet) or networks experiencing measurable packet loss. Assign a value between 0 and 5 to append that number of previously sent packets onto the current packet. Values larger than 2 should be assigned for networks experiencing packet loss greater than one percent.
fixed-packets	Enable/disable the addition of *n*-length payload pairs at the end of the packet when packet-redundancy is enabled, where *n*- is the value of the packet-redundancy parameter.

Parameter	Setting
max-data-rate	Assigns a maximum rate in bits per second (bps) to the real-time fax transmission. Assign either 2400, 4800, 9600, or 14400.

In an SS7 environment, values in IPDC messages override corresponding call management settings in the default Voip profile. For information about IPDC support for real-time fax, see "IPDC message support for real-time fax and transparent modem" on page 5-8.

The following example shows the commands that enable T.38 fax call processing and leave all performance parameters enabled:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set rt-fax-options rt-fax-enable = yes
admin> write
VOIP/{ 0 0 } written
```

Routing fax calls

Once real-time fax is enabled on a TAOS unit, the MultiVoice administrator needs to create the appropriate coverage areas on the MVAM to ensure that fax calls are routed between gateways that are fax capable.

Both the originating and connecting Gateway must be capable of detecting the CED tone or the V.21 HDLC flag, and acknowledging (respond with the proper ACK signal) for fax transmission to succeed. Only MultiVoice Gateways that have been loaded with the proper licensed software code and have the real-time fax feature enabled have this capability.

To ensure proper routing and processing of fax calls, the MultiVoice administrator can do the following:

- Enable real-time fax on all gateways. This ensures that any fax call which comes in can be connected regardless of routing. This requires that all gateways used by that network are TAOS units.
- Enable real-time fax on selected gateways, and assign them to a virtual zone managed by MVAM. This solution uses a feature available from the MultiVoice Access Manager which allows it to administer the fax gateways as if they were part of a separate H.323 zone. Access numbers for these gateways could be reserved for fax calls only. Calls initiated and completed within the virtual zone would always be between fax capable gateways. Voice calls could still be initiated and completed both within that virtual zone and across any other zones administered by MVAM. This reduces the number of TAOS units necessary to support network fax operations.

Using error correction mode (ECM)

The ecm-enable parameter enables/disables error correction mode (ECM) for real-time fax calls. Change the setting from its default value to no, when performing real-time fax processing on MultiVoice networks with low packet loss and low latency characteristics.

When this parameter is set to yes (the default), fax frames are retransmitted whenever a frame is not received correctly. ECM frames are relayed end to end between terminals. For certain network environments, using the default value for the `ecm-enable` parameter might result in the following:

- Fax transmission delays
- Diminishment fax call performance
- Disconnection of fax calls in progress
- Fax calls time out without ever connecting

Enabling packet redundancy

A packet redundancy scheme and jitter buffer improve the performance of MultiVoice real-time fax over unmanaged networks (such as the public Internet). Packet redundancy allows the MultiVoice Gateway to process several hundred milleseconds of packet jitter and allows the optional transmission of redundant packet data for fax calls across networks experiencing instances of packet loss and packet jitter.

To support this feature, MultiVoice requires real-time fax to be enabled on the MultiVoice Gateway. This may be verified by checking the base profile for the `rt-fax-enabled=yes` entry.

Packet redundancy parameter

Redundant packet data is defined as the last *n* packets transmitted appended to the current packet. The value of *n* is set through the CLI using the `packet-redundancy` parameter in the `rt-fax-options` sub-profile of the `voip` profile.

Assigning the `packet-redundancy` parameter a value (such as packet-redundancy = 4), causes the TAOS unit to append the specified number of previously sent packets onto the current packet. On networks experiencing measurable packet loss, this improves the reliability of the fax transmission.

Depending upon the amount of measurable packet loss for a network, the redundancy parameter should be set accordingly:

Network condition	**Recommended value(s)**
Packet loss occurs in frequent bursts.	1 through 5
Occasional packet loss (less than one percent)	0 (default)
Occasional packet loss (greater than one percent)	1 or 2

The additional bandwidth required for each fax call increases proportionally to the level of redundancy, adding 50 bytes of packet data per increment.

Packet redundancy uses a slip buffer to

- Allow MultiVoice real-time fax to tolerate packet jitter
- Keep the modem fed with data, preventing modem underrun

S 1513

Fixed-size packet format

The packet redundancy scheme uses a fixed-size packet format, consisting of a 49-byte payload, a prefixed sequence number, and a length field that preceeds the payload data. When packet redundancy is enabled, *n*-length payload pairs are added at the end of the packet; where *n* is the value of the packet-redundancy parameter.

Previously, the TAOS unit sent variable length packets that were guaranteed to be zero terminated; allowing Class 1 modems to underrun gracefully.

The packet-redundancy parameter causes the TAOS unit to append the designated number of previously sent fax packets onto the current packet. On networks experiencing measurable packet loss, this improves the reliability of the fax transmission.

This parameter accepts values from 0 through 5, directing MultiVoice to append the designated number of previously transmitted fax packets to the current packet, as follows:

Parameter value	Specifies
0	No change from the default packet behavior.
1	Append and send the previous fax packet with the current fax packet.
2	Append and send the two previous fax packets with the current fax packet.
3	Append and send the three previous fax packets with the current fax packet.
4	Append and send the four previous fax packets with the current fax packet.
5	Append and send the five previous fax packets with the current fax packet.

The following example illustrates how to change the default value of the packet-redundancy parameter.

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set packet-redundancy=4
admin> write
VOIP/{ 0 0 } written
```

The following dependencies apply to this parameter:

- Once saved, packet redundancy is enabled with the next VoIP call
- This value is set to N/A when fixed-packets=no.

Disabling Packet Redundancy

The fixed-packets parameter enables MultiVoice Gateways running TAOS 9.0 to process real-time fax calls to and from MultiVoice Gateways running earlier TAOS versions. The new packet sequence numbering introduced in TAOS 10.0 for real-time

fax required a format change, creating high speed data packets. When these packets are absent (for example, a fax call is initiated from a MultiVoice Gateway running a pre-TAOS 10.0) MultiVoice Gateway interprets image data as sequence data. Also the smaller packets forwarded by the new code rely on the slip buffer to keep the modem fed with data or it drops the carrier.

The `fixed-packets` parameter disables use of redundant packets and the slip buffer, enabling the gateway to use pre-TAO 10.0 fax packet scheme.

When enabled, fax calls are processed using variable length packets that are zero terminated; allowing Class 1 modems to underrun gracefully.

When the value of this parameter is yes, the default, the pre-TAOS 10.0 fax packet scheme is enabled. When the value of this parameter is no, jitter buffering and packet redundancy for real-time fax processing is disabled. Once saved, the selected packet scheme is enabled with the next fax call.

The following example illustrates how to disable jitter buffering and packet redundancy on a TAOS unit:

```
admin> read voip { 0 0}
VOIP/{ 0 0} read
admin> set fixed-packets=no
admin> write
VOIP/{ 0 0 } written
```

The following dependency applies to this parameter:

- When this value is set to yes, then `packet-redundancy=n/a`.

Verifying H.323 fax call operations

The h323debug command verifies fax call operations. This diagnostic command reports when a fax session

- Is initiated
- Is acknowledged (ACK)
- Fails

For each event, h323debug reports the following

Table 5-2. Events reported by the h323debug command (Page 1 of 2)

Event	**Message**	**Condition**
Fax session is successfully initiated on the near-end gateway	H323: *<CALL#>*[a]: stackH245Event: Received mode request for T38	The gateway detected a CED tone or the V.21 HDLC flag received from the PSTN/gateway.
Fax session request is successfully received on a far-end gateway	H323: *<CALL#>*[a]: stackH245Event: Received mode request ACK for T38	The gateway acknowledged a CED tone or the V.21 HDLC flag received from the connecting gateway.

S 1515

Table 5-2. Events reported by the h323debug command (Page 2 of 2)

Event	**Message**	**Condition**
Fax session fails to initiate on the near-end gateway	H323: *<CALL#>*[a]: WARNING: _stackH245Event() FAX not enabled, request mode refused!	The gateway could not respond to a CED tone or the V.21 HDLC flag received from the PSTN/gateway.

a. *<CALL#>* is the unique identifier assigned to a call by the originating Gateway.

The h323ShowStats command provides the following summary of fax call activity, for failed fax requests:

```
FAX  Originate Failed        = 0
```

IPDC message support for real-time fax and transparent modem

IPDC message request packet pass-through call (RCCP), accept packet pass-through call (ACCP), request modify for packet pass-through call (RMCP), and accept modify packet pass-through call (AMCP) messages enable an Signaling Gateway (SoftSwitch) to direct the TAOS unit to enter T.38 fax mode or transparent modem mode on the basis of tone detection.

In addition, the Signaling Gateway (SoftSwitch) can control echo cancellation by disabling it or setting it to 32 milliseconds on a per-call basis.

Transparent data

Transparent data enables users to run a modem on an SS7 VoIP channel using IPDC, regardless of the vocoder in use. Transparent data is encoded as an audio-mode type, either G.711 μ-law (64K) or G.711 a-law (64K).

MultiVoice Gateways detect fax/modem tones in both the TDM connection and RTP stream. When fax/modem tones are detected, echo cancellation and suppression are automatically disabled. When codes other than the G.711 μ-law and a-law are used, IPDC messages allow the SoftSwitch to request the TAOS unit to enable G.711 transparent data mode upon fax tone detection from the MultiVoice Gateway.

T.38 fax

Real-time fax (if supported) is encoded as a data-mode type, T.38 fax. T.38 fax is used to carry facsimile traffic over an IP link. Currently, T.38 is only employed over H.323 VoIP calls. New IPDC messages allow the SoftSwitch to tell the TAOS unit to enter T.38 fax mode upon fax tone detection from the MultiVoice Gateway.

Note The real-time fax license is required to enable T.38 fax for IPDC.

Echo canceller

The Echo Cancellation Tag (0x74) is implemented on a per-call basis. Only values of 0 (off) and 32 milliseconds are currently supported.

An echo canceller, compliant with International Telecommunications Union's (ITU) G.168-2000 standard is supported for the G.711 and G.729A audio codecs.

Echo occurs when a speaker's speech signal is coupled into the receive path from the far end. If the echoed signal has sufficient amplitude and delay, the speaker may experience annoying echo. The primary cause of the returned echo signals is the *hybrid*, which performs the necessary 4-wire to 2-wire conversion between the 4-wire facilities of the telecommunications network and the 2-wire telephone circuit.

ITU G.168-2000 Recommendation

The ITU's G.168-2000 Recommendation defines objective tests, that if passed, will ensure a minimum level of quality within the network. This recommendation increases the scope of the tests defined in G.165 and ensures that echo canceller performance is adequate under wider network conditions, such as performance on voice, FAX, residual acoustic echo signals, and mobile networks.

Lucent Technologies' echo canceller meets or exceeds all of the objective tests defined in the G.168-2000 Recommendation. Additionally, several subjective evaluations have been performed to ensure the highest possible performance and robustness.

The new echo canceller provides 64ms echo tail cancellation for the G.711 audio codec and 32ms echo tail cancellation for the G.729A audio codec in order to properly model and cancel the echo from severe hybrid impedance mismatch. All other voice codecs (for example, G.728, G.723.1, Full-Rate GSM) use the ITU-G.165 standard.

Notify tone (NTN)

The notify tone (NTN) message notifies the Signaling Gateway (SoftSwitch) when an asynchronous fax or modem tone is detected. The TAOS unit sends this message to the Signaling Gateway (SoftSwitch) if either fax or modem tone detection is enabled and the Signaling Gateway (SoftSwitch) identifies the tone. The TAOS unit detects fax tone if `rt-fax-enable` is set to yes in the default `voip` profile or if it receives the relevant IPDC message from the Signaling Gateway (SoftSwitch).

The TAOS unit detects modem tone if `g711-transparent-data` is set to yes in the default `voip` profile or if it receives the relevant IPDC message from the Signaling Gateway (SoftSwitch).

Changes to existing message tags

The following existing message tag values are modified for the NTN message to support T.38 and transparent modem/fax detection.

Table 5-3. Modified NTN message tag values (Page 1 of 2)

Tag	Description	Values
0x33	Tone string	f: Fax tone (CED, no phase reversal, or V.21 flags)
		o: Modem tone (CED, phase reversal)

Table 5-3. Modified NTN message tag values (Page 2 of 2)

Tag	Description	Values
0x49	Tone type	0x06: Fax tone (CED, no phase reversal, or V.21 flags)
		0x07: Modem tone (CED, phase reversal)

The following existing message tag values are modified for the RCCP, ACCP, RMCP, and AMCP messages to support T.38 and transparent modem and fax detection.

Table 5-4. Modified RCCP, ACCP, RMCP and AMCP message tag values

Tag	Description	Values
0x70	Encoding type	0x60: Transparent Data encoding[a]
		0x61: T.38 Fax over UDP

a. Currently, Transparent Data is G.711 RTP with several features disabled.

RCCP, ACCP, RMCP, and AMCP message tags

The following new message tag values are added to the RCCP, ACCP, RMCP and AMCP messages to support T.38 and transparent modem/fax detection. These values are applied on an individual call basis.

Table 5-5. New the RCCP, ACCP, RMCP and AMCP *message tag values*

Tag	Description	Values
0x74	Echo Cancellation	0x00 : Echo canceller off (0 msecs.)
		0x01 : Echo canceller on (32 msecs.)
0x77	Constant Fax tone detection	Report which fax tone support is enabled (either `rt-fax-enable=yes` or `g711-trans-data=yes`) and override this setting if appropriate.
0x78	Constant Modem tone detection	Report whether modem tone support is enabled (either `g711-trans-data=yes` or `g711-trans-data=no`) and override this setting if appropriate.

New NTN message tags

The NTN message is sent by the MultiVoice Gateway to Signaling Gateway (SoftSwitch) when fax or modem tone detection is enabled and either tone is detected. The fax/modem tone detection can be enabled or disabled either by the IPDC tags in RCCP and RMCP messages or in the `voip` profile.

S 1518

The following new NTN message is added to support T.38 and transparent modem/fax detection. These values are applied on an individual call basis.

Table 5-6. New NTN message

Tag	Description	Values
0x00F0	Notify Tone	This NTN message from the TAOS unit notifies Signaling Gateway (SoftSwitch) of asynchronous fax/modem tone detections.

The following tags may be included in a Notify Tone message:

Table 5-7. Notify Tone message tags

Tag	Description	Values
0x65	Source Post Type	Required.
0x07	Source Module Number	Required.
0x0D	Source Line Number	Required.
0x15	Source Channeled Number	Required.
0x40	Ascend Route ID	Optional.
0x33	Tone String	Required.
0x49	Tone Type[a]	Required.

a. Currently Tone String and Tone Type convey the same information.

ss7nmi command

The ss7nm -m command output includes modifications for the RMCP, AMCP and NTN messages as illustrated by the following:

```
admin>ss7nmi -m
IPDC message processing statistics:

  Message code       Received      Sent
  RCR    (0x0011):          1         0
  ACR    (0x0012):          0         1
  RCCP   (0x0013):          1         0
  ACCP   (0x0014):          0         1
  RMCP   (0x0015):          1         0
  AMCP   (0x0016):          0         1
  RMS    (0x0041):          1         0
  NMS    (0x0042):          0        17
```

```
RLS   (0x0043):    1    0
NLS   (0x0044):    0    1
NSUP  (0x0081):    0    1
ASUP  (0x0082):    1    0
NTN   (0x00f0):    0    1

Data collection was started: [04/26/2000 15:40:47]
```

Max Data Transmission Rate Configuration

The `max-data-rate` parameter in the `rt-fax-options` sub-profile of the `voip` profile sets the maximum data transmission rate allowed for a T.38 fax session configurable on a MultiVoice Gateway. The bandwidth used for fax sessions on your networks can be regulated.

Through the MultiVoice Gateway administration interface, you can modify the rate negotiation between the originating and destination fax terminals. This improves the reliability of the fax transmission by selecting lower fax transmission rates, resulting in fewer lost or repeated fax packets and requiring less bandwidth for fax transmissions.

Rate modification of the fax transmission rate is accomplished by modifying the content of the Digital Identification Signal (DIS) frame transmitted from the destination fax, using the `max-data-rate` parameter in the `rt-fax-options` sub-profile. Upon receipt of that DIS frame, the originating fax will use the data transmission rate specified in the Max Rate parameter (or slower), and a supported modulation type. The content of the DIS frame is defined in the ITU Telecommunication sector standard (ITU-T) T.30, *Procedures for document facsimile transmission in general switched telephone networks.*

Changing the `max-data-rate` parameter modifies the high-speed data transmission rate reported by the destination fax, and masks certain modulation types associated with higher fax transmission speeds. For example, once the data rate is set for 9600 bps, V.17 and V.33 are disallowed even though V.17 supports 9600 and 7200 bps. This is necessary since the DIS frame can specify only the supported modulation types for the highest selected transmission speeds on the destination fax, and since the calling fax terminal requires “training” to match the supported modulation. The value assigned to the Max-Rate parameter on the egress MultiVoice Gateway sets the maximum fax transmission rate for the call.

Setting the max-data-rate parameter

The `max-data-rate` parameter modifies the rate negotiation between the originating and destination fax terminals. This improves the reliability of the fax transmission by reducing the number of lost or repeated packets which occur during high rate transmissions, and reduces the required bandwidth for fax transmissions. Changes made to this parameter setting take effect with the next VoIP call.

Values assigned to this parameter cause MultiVoice to do the following:

Parameter value	Specifies
14400	(Default). Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 14,400bps.
9600	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 9,600bps.
4800	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 4,800bps.
2400	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 2,400bps.

The following example illustrates how to set the fax data transmission rates:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> list rt-fax-options
[in VOIP/{ 0 0 }:rt-fax-options]

admin> set max-data-rate=9600

admin> write
VOIP/{ 0 0 } written
```

This parameter has the following dependency:

- This parameter is N/A when rt-fax-enable=no.

Fax session detection

In a regular fax session, the called terminal (fax machine) sends a CED tone (a continuous 2100Hz +/-15Hz, -10 dBm nominal, for not less than 2.6 seconds and not more than 4 seconds) to indicate a connection with a called nonspeech terminal. Then CCITT V.21 channel #2 (FSK) modulation at 300BPS from the called terminal begins. Each FSK frame is preceded by one second (+/- 15%) of HDLC flags 0x7E. The one and zero correspond to the tones at 1650Hz and 1850Hz (lasting a bit duration, 1/300 sec) respectively in FSK frames. Since both signals are generated by the called terminal, they should be detected on the egress gateway. Fax call processing is initiated as follows:

```
Call -> Answer -> CED -> FSK Frames ->
```

Because certain fax machines do not send CED, both the CED and the V.21 flags detection is implemented on the egress gateway to detect fax calls. Once either tone is detected, the slave DSP informs i960 through SARM. When CED is detected, the

S 1521

gateway initiates the switch over to a fax call without waiting on detection of V.21 flags. Then the fax session begins. The detection ranges are as follows:

Table 5-8. Detection ranges

Detected tones	frequency tolerance	min reception level
CED	+/- 21Hz	-30dBm
V.21 Flags :	+/- 21Hz	-27dBm

Fax compatibility

While real-time fax is generally compatible with any fax machine with supports CED and V.21 signaling, the following faxes/fax software have been successfully tested with MultiVoice real-time fax:

- Ricoh Fax 7700L
- Brother Intellifax 770
- Canon Faxphone B640
- Xerox Workcenter 450c
- Sharp Ux107
- Brother Intellifax I550
- Panasonic Kx-FM220
- Winfax Pro
- Procomm Plus
- Telegra scripts

Note The Mitsubishi FA 1500W, which does not retransmit the DCS/TCF exchange during the negotiation phase of a fax call, proved to be incompatible with MultiVoice real-time fax.

Network Reporting

6

Network reporting options

When a TAOS unit is used as a MultiVoice Gateway, the unit can be configured to report call activity using either Simple Network Management Protocol (SNMP) or by generating call record information for NavisAccess™.

Base profile parameters

A TAOS unit must be licensed to support extended network management reporting of VoIP and fax. Extended network management reporting is enabled when the `network-mgmt-voip-enabled` parameter is present and set to yes. This allows monitoring of MultiVoice using SNMP or NavisAccess™.

Enabling SNMP traps for MultiVoice

VoIP-enabled TAOS units can generate notifications for the following MultiVoice Gateway events:

- Change in the call logging server
- Change in configured Gatekeeper for VoIP
- Change in state of a WAN line

For the traps to be sent, they must be enabled in the system and the individual trap conditions must be set to yes. For details about enabling traps, see the *APX/MAX TNT Administration Guide.*

S 1523

Following are the relevant parameters (shown with default values) for enabling the individual trap conditions:

```
[in TRAP/""]
call-log-serv-change-enabled = no
voip-gk-change-enabled = no
wan-line-state-change-enabled = no
```

Parameter	Setting
call-log-serv-change-enabled	Enable/disable notification when the call-logging server changes. If the call-logging server index is changed or if the IP address of the active call-logging server is changed, this trap notification sends the following information to the SNMP manager: • The new call logging server index (callLoggingServerIndex) • The IP address of new call logging server (callLoggingServerIPAddress) • The absolute that the server change occurred (sysAbsoluteCurrentTime) (Ascend Trap 38)
voip-gK-change-enabled	Enable/disable notification when the registered gatekeeper changes. If a new Gatekeeper is registered with the gateway, a register request (RRQ) message is sent from the gateway to the new gatekeeper. When the gateway receives the admission request (ARQ) message from the new gatekeeper, this notification sends the following information to the SNMP manager: • The new gatekeeper index (voipCfgGkIndex) • The IP address of new gatekeeper (voipCfgGkIpAddress) • The absolute time that the gatekeeper change occurred (sysAbsoluteCurrentTime) (Ascend Trap 39)
wan-line-state-change-enabled	Enable/disable notification if the state of an E1 or T1 line changes. This trap sends the following information to the SNMP manager: • The T1 or E1 line interface index (wanLineIfInde) • The line usage (wanLineUsage). This usage is reported as trunk, quiesced, or disabled. • The absolute time that the line state changed (sysAbsoluteCurrentTime) (Ascend Trap 40)

The VoIP MIB (ascend 28)

The VoIP MIB enables network management stations to monitor MultiVoice Gateway operations using SNMP. Attributes in the MIB can be obtained by SNMP Get and Get-Next commands. The MIB uses the following object identifiers for identifying MultiVoice Gateway or MultiVoice Gatekeepers to a network manager:

- voipCfgGroup (voipGroup 1)
- voipCfgGkGroup (voipCfgGroup 1)
- voipCfgGwGroup (voipCfgGroup 2)

The MIB uses the following tables for identifying MultiVoice Gateway and MultiVoice Access Manager functions.

```
voipCfgGkTable OBJECT-TYPE (voipCfgGkGroup 1)
   SYNTAX SEQUENCE OF VoipCfgGkEntry
   ACCESS not-accessible
   STATUS mandatory
   DESCRIPTION A list of entries for H323 network Gatekeeper.
voipCfgGkEntry OBJECT-TYPE (voipCfgGkTable 1)
   SYNTAX VoipCfgGkEntry
   ACCESS not-accessible
   STATUS mandatory
   DESCRIPTION An entry holding information about the Gatekeeper for
   the system.
   INDEX (voipCfgGkIndex)
VoipCfgGkEntry:
   SEQUENCE :
      voipCfgGkIndex-INTEGER
      voipCfgGkStatus-INTEGER
      voipCfgGkIpAddress-IpAddress)
voipCfgGkIndex OBJECT-TYPE ( voipCfgGkEntry 1)
   SYNTAX INTEGER
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION This number uniquely identifies the Gatekeeper.
voipCfgGkStatus OBJECT-TYPE (voipCfgGkEntry 2)
   SYNTAX INTEGER:
      registered(1)
      not_registered(2)
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION This value indicates whether the gateway is registered
   with the Gatekeeper.
voipCfgGkIpAddress OBJECT-TYPE (voipCfgGkEntry 3)
   SYNTAX IpAddress
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION The IP address of the Gatekeeper.
voipCfgGwVpnMode OBJECT-TYPE (voipCfgGwGroup 1)
   SYNTAX INTEGER:
      no (1)
      yes(2)
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION Virtual Private Network Toggle Switch.
```

```
voipCfgGwPktAudioMode OBJECT-TYPE (voipCfgGwGroup 2)
   SYNTAX INTEGER:
      other(1)
      g711_ulaw(2)
      g711_alaw(3)
      g723(4)
      g729(5)
      g723_6_4kps(6)
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION Audio Coder to be used for voice packetization.
```

The voipCfgGwVpnMode and voipCfgGwPktAudioMode objects can be accessed using index 0 because they are separate leaves in the MIB tree.

The voipCfgGkIndex, voipCfgGkCurrent and voipCfgGkIpAddress objects are located in the voipCfgGkTable table. They can be obtained using voipCfgGkIndex as an index.

Sending H.323 call information to SNMP log clients

H.323 call information from MultiVoice Gateways can be collected. This includes the capability to generate start, stop, and call progress records for both VoIP and fax calls.

H.323 call information from MultiVoice Gateways performing VoIP call processing can be sent to SNMP log clients. Each MultiVoice Gateway provides the following H.323 call information:

- Billing start records
- Billing stop records
- Call disconnect records
- Fax start records

Billing start records

A billing start record reports the point in the call where speech communications is established. Start records provide the following information:

Attribute	Specifies
Call ID	The H.323 protocol call ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call, and is used to track all call processing events related to a particular call.
Conference ID	The H.225.0 protocol messaging ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call, and is used to track H.225.0 call setup messages related to a particular call.
Dial time	The time a user spends dialing the destination telephone number. This is the time a MultiVoice Gateway waits to collect the dialed telephone number. This value is zero for calls originating from the LAN.

Attribute	Specifies
`Setup time`	The time from the moment a user finishes dialing the destination telephone number until the moment the speech is established to the called destination.
`Call origin`	The IP address used to identify the calling origin. This can be the ingress MultiVoice Gateway or an H.323-compliment terminal.
`Remote IP`	The IP address used to identify the called destination. This can be the egress MultiVoice Gateway or an H.323-compliment terminal (PC).
`Telephone number`	The dialed number string entered by the user.
`CLID number`	The E.164 address associated with the calling origin.
`Audio mode`	The audio codec used to connect an H.323 call.

Billing stop records

A billing stop record reports the point in the call where speech communications terminates (end points go on-hook). Stop records provide the following information:

Attribute	Specifies
`Call ID`	The H.323 protocol call ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call, and is used to track all call processing events related to a particular call.
`Conference ID`	The H.225.0 protocol messaging ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call and is used to track H.225.0 call setup messages related to a particular call.
`Connect time`	The time from the moment speech is established until the callers hang up (go onhook) normally.
`Drop time`	The time a call connection is dropped by the WAN or LAN connection, which ever signal is reported first.
`Drop reason`	The H.323 call drop reason. For normal call termination, the billing stop record reports `normalDrop`.

Call disconnect records

A call disconnect record is generated whenever a call is not terminated normally (such as when a connection between end points is lost as a result of equipment failure or network failure). Disconnect records provide the following information:

though some information may not be present as depending upon the origin of the call failure:

Attribute	Specifies
`Call ID`	The H.323 protocol call ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call, and is used to track all call processing events related to a particular call.
`Conference ID`	The H.225.0 protocol messaging ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call, and is used to track H.225.0 call setup messages related to a particular call.
`Dial time`	The time a user spends dialing the destination telephone number. This is the time a MultiVoice Gateway waits to collect the dialed telephone number. This value is zero for calls originating from the LAN.
`Setup time`	The time from the moment a user finishes dialing the destination telephone number until the moment the speech is established to the called destination.
`Call origin`	The IP address used to identify the calling origin. This can be the ingress MultiVoice Gateway or an H.323-compliment terminal.
`Remote IP`	The IP address used to identify the called destination. This can be the egress MultiVoice Gateway or an H.323-compliment terminal (PC).
`Telephone number`	The dialed number string entered by the user.
`CLID number`	The E.164 address associated with the calling origin.
`Audio mode`	The audio codec used to connect an H.323 call.
`Drop from`	The location which disconnected the call, either WAN or LAN.
`Drop reason`	The H.323 call drop reason. For disconnect reports, this is an incomplete and interrupted call termination reason.

Fax start records

A fax start record is generated whenever a fax answer tone is detected during a VoIP. The fax record provides the following information:

Attribute	**Specifies**
`Call ID`	The H.323 protocol call ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call, and is used to track all call processing events related to a particular call.
`Conference ID`	The H.225.0 protocol messaging ID. This is a unique nonzero number assigned by a MultiVoice Gateway upon receipt of a call and is used to track H.225.0 call setup messages related to a particular call.
`Modulation type`	The fax modulation type detected by the MultiVoice Gateway (such as V.21, V.27, V.29, V.17, etc.)
Speed	The transmission speed, modulation rate, detected for this fax transmission by the MultiVoice Gateway (such as 2400, 4800, 7200, etc.)

Note Fax records are generated for T.38 fax transmissions.

H.323 disconnect reasons

H.323 disconnect reasons have been added to `disconnect-reason-type.mibdef` for the Ascend disconnect type. Reported disconnect reasons for standard and nonstandard call termination are recorded in the following table.

Call drop reason	**Call drop code**	**Specifies**
DIS_H323_DROP_REASON_NULL	500	Call drop reason not available
DIS_H323_DROP_REASON_NORMAL	501	Normal disconnect (caller hung up)
DIS_H323_DROP_REASON_DEST_BUSY	502	Called destination busy
DIS_H323_DROP_REASON_DEST_UNREACHABLE	503	Called destination unreachable
DIS_H323_DROP_REASON_REJECT	504	Call rejected by TAOS
DIS_H323_DROP_REASON_WAN_FAILURE	505	WAN failure, egress MultiVoice Gateway could not connect the call

Call drop reason	Call drop code	Specifies
DIS_H323_DROP_REASON_GATEWAY_RESOURCES	506	Egress MultiVoice Gateway could not process the call
DIS_H323_DROP_REASON_NO_BANDWIDTH	507	Sufficient bandwidth not available on the WAN for this call
DIS_H323_DROP_REASON_GW_NOT_REGISTERED	508	Egress MultiVoice Gateway is currently unregistered with the MVAM
DIS_H323_DROP_REASON_INVALID_PIN	509	Caller entered an invalid PIN
DIS_H323_DROP_REASON_INVALID_DNIS	510	Caller dialed invalid number for called destination
DIS_H323_DROP_REASON_NO_LAN_ANSWER	511	A LAN connection was not available
DIS_H323_DROP_REASON_STATE_MACHINE	512	Call state machine on MultiVoice Gateway could not advance
DIS_H323_DROP_REASON_NO_LAN_DISCONNECT	513	The WAN dropped the connection
DIS_H323_DROP_REASON_FEGW_CAUSE_CODE	514	The egress MultiVoice Gateway dropped the connection
DIS_H323_DROP_REASON_MAX_PIN_ATTEMPTS	515	The caller failed to authenticate on all attempts to enter the PIN
DIS_H323_DROP_REASON_CODER_DENIED	516	The MultiVoice Gateway could not negotiate an audio codec selection with the far-end gateway

NavisAccess™ support for VoIP call reporting

Basic VoIP call reporting using NavisAccess™ includes the capability to generate start records, stop records, and call progress records for both VoIP and fax calls. These